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(54) Titre : SYSTEME ET METHODE DE MESURE DES EMISSIONS OTO-ACOUSTIQUES DE DISTORSION (DPOAE)
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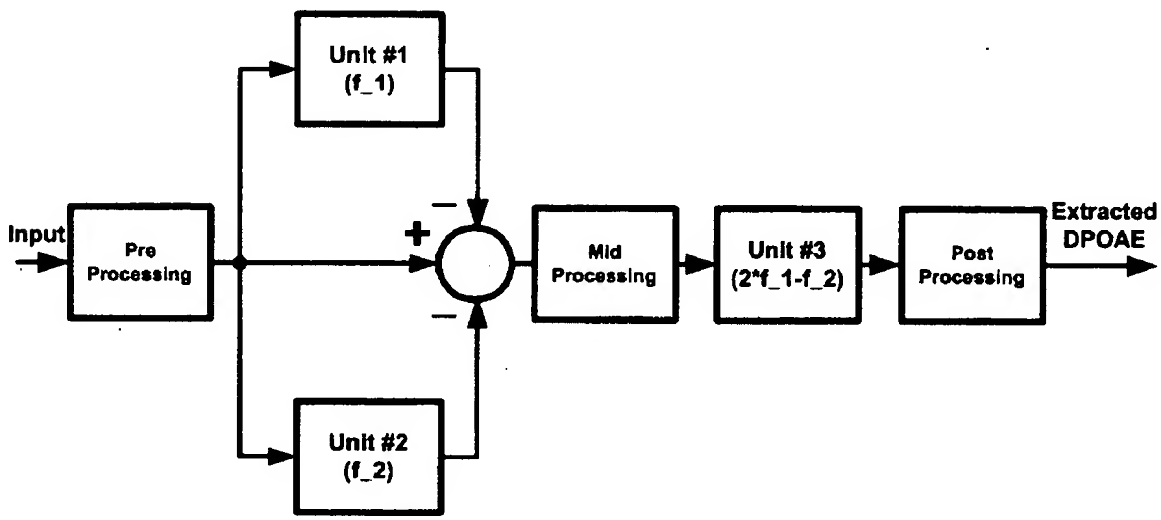

(57) Abrégé/Abstract:
A new system and method for the measurement of DPOAE signal level is presented. The present invention is based on a recently developed nonlinear adaptive algorithm. Three units of such algorithms are employed to extract and measure the two artifacts and the DPOAE signal itself. Each core unit has the capability of locking to a specified sinusoidal component of its input signal and tracking its variations over time. The present method features structural simplicity which renders it suitable for both software and hardware implementation. It also offers a high degree of noise immunity which is useful in clinical examinations. Finally, compared to conventional methods the present method requires shorter measurement time.

DPOAE MEASUREMENT SYSTEM AND METHOD

Abstract

A new system and method for the measurement of DPOAE signal level is presented. The present invention is based on a recently developed nonlinear adaptive algorithm. Three units of such algorithms are employed to extract and measure the two artifacts and the DPOAE signal itself. Each core unit has the capability of locking to a specified sinusoidal component of its input signal and tracking its variations over time. The present method features structural simplicity which renders it suitable for both software and hardware implementation. It also offers a high degree of noise immunity which is useful in clinical examinations. Finally, compared to conventional methods the present method requires shorter measurement time.

Introduction

Distortion product otoacoustic emissions (DPOAEs) are very low level stimulated acoustic responses to two pure tones presented to the ear canal. DPOAE measurement provides an objective non-invasive measure of peripheral auditory function and is used for hearing assessment. DPOAE screening is becoming a standard clinical practice to predict potential sensorineural hearing loss especially in newborns.

DPOAEs have been recognized for a number years. However, DPOAE measurement is considered an active area of research because of the challenging nature of the signal processing task. To address the ever-increasing demand for high performance DPOAE measurement methods, a number of signal processing algorithms have been presented in recent years. With the availability of the powerful computational tools such as digital signal processors (DSPs), commercial medical devices dedicated to DPOAE measurement are becoming available.

In this type of otoacoustic test, two pure tones with frequencies f_1 and f_2 are presented to the cochlea. For best results, f_2 is usually chosen as $1.2f_1$. Due to the non-linearity of the ear, a very low level distortion product of frequency $2f_1 - f_2$ is generated in normal ears. The level of such a DPOAE signal is a measure of the functionality of the ear. Estimation of such a weak signal buried under two strong artifacts in a potentially noisy background is a challenging signal processing problem.

Conventionally, fast Fourier transform (FFT) is used as the main signal processing tool to estimate the level of the DPOAE signals. Application of FFT in this problem has a number of shortcomings among which long measurement time is most pronounced. Such a long measurement time is usually required for acquisition of more data which, when averaged, reduce the overall background noise effect. Unreliability of the measurements is another

problem of FFT-based methods and is a direct effect of the sensitivity of the FFT-based methods to the background noise. In addition to the need to increase the measurement time, the tests are usually required to be conducted in low noise environments such as sound-proof booths.

In an attempt to devise high performance DPOAE estimation techniques, linear adaptive signal processing techniques have been employed. Such techniques generally offer better performance in terms of measurement time which may be interpreted as higher noise immunity of adaptive techniques compared to FFT. However, the need for sound-proof examination rooms is not obviated with such techniques.

This document presents a system and a method of measurement of DPOAE signals based on a new signal processing technique in which individual sinusoidal components of a given signal are extracted and their variations are adaptively tracked over time. The present DPOAE estimation method employs three units of core algorithms which are described in detail in pending Canadian patent entitled "System and Method of Extraction of Sinusoids of Time-varying Characteristics" filed on May 28th, 2001, by Alireza Karimi Ziarani, serial number: 2,349,041.

The two artifacts are first extracted by two units and are subtracted from the input signal to generate an input signal of which DPOAE signal has a higher relative portion. Such a signal is then fed to another core unit which estimates the level of DPOAE signal. Superior performance of the present technique in terms of noise immunity and fast measurement is illustrated with the aid of computer simulations.

In order to provide a brief background on the subject, a general review of the structure of a generic DPOAE measurement device is presented in the next section. It will be seen that the heart of such an apparatus is the signal processing subsystem which, in the final

analysis, determines the performance of the overall system. Later in this document the structure of the present invention for the signal processing module is presented and its performance is demonstrated.

Structure of a Generic DPOAE Measurement Device

In this section a brief overview of the structure of a typical DPOAE detection system is provided. Figure 1 shows the generic block diagram of a DPOAE measurement device. It consists of three main modules: the data acquisition/transducers module, the signal processing module and the display.

Data acquisition unit is the medium between the processing unit and the probe which transmits and receives acoustic signals in the audio range. Components of the compound data acquisition/transducers module are illustrated in more detail in Figure 2. One of the main functions of this module is to convert digital signals produced by the signal processing module to analog signals which are then conditioned and converted to audio signals. Conditioning of the signals in this case may or may not include filtering. Conversely, the audio signals recorded by the probe are conditioned and converted to digital signals to be processed by the signal processing module.

The signal processing module is the heart of the system which produces the digital form of the artifacts and extracts and measures the DPOAE signal. A DSP, or if the computational/architectural demand is low even a microcontroller, can be employed as the hardware platform of this unit. Signal processing is embedded as the software in such a hardware platform. Alternatively, and provided that the complexity of the signal processing algorithms remains low, signal processing unit may be implemented solely in hardware using programmable logic array (PLA) or field programmable gate array (FPGA) tech-

nology. In an ideal case, namely when the signal processing algorithm is not excessively complex, the hardware does not require a PC for its operation; however, interfacing to a PC is usually provisioned for data management.

The display unit is the interface between the device and the operator. It can be a simple LED/LCD and/or a small printer.

Present Technique

Figure 3 shows the main functions of the software embedded in the signal processing module. The software is essentially responsible for the generation of the artifact signals and extraction of DPOAE as well as management of input/output data. As discussed before, the significance of the present invention is in the introduction of a signal processing technique for the extraction and measurement of DPOAE signals.

The present signal processing scheme employs three core units to construct a high performance DPOAE extraction module. Each core unit is capable of focusing on and extracting a pre-specified sinusoidal component of its input signal which may contain many other components including noise. More importantly, they can effectively follow variations in the amplitude, phase (and frequency) of the extracted sinusoidal component. Although the underlying mathematics ensuring stability and performance of such core units is very complex, the structure of the core units remain extremely simple. They are found to be very robust with respect to variations in the internal settings as well as external conditions and exhibit superior performance over existing linear adaptive and FFT-based algorithms.

The input signal is often assumed to consist of two pure sinusoids with frequencies f_1 and f_2 at a very high level (usually about 60 to 70 dB) and a very low level DPOAE $2f_1 - f_2$

at about -15 to 15 dB. It is polluted by a noise usually considered to be at about -10 to 10 dB level. The noise in fact represents the totality of all undesirable signals that may be present in the environment in which the examination is being conducted as well as unavoidable white Gaussian noise. Because of excessive degree of pollution (artifacts and noise), one single core unit set to extract the DPOAE signal out of the input exhibits poor performance. Different arrangements were studied to construct a high performance architecture. One of the most successful configurations is shown in Figure 4. Three core units are employed. The first two core units are set to extract the artifacts. They effectively do so with very small errors. The extracted artifacts are then subtracted from the input to produce a signal, of which DPOAE has a higher relative portion. The third core unit is then set to extract DPOAE.

The structure of the main embodiment of the present invention is essentially based on the concept of extraction of artifacts and the DPOAE as illustrated in Figure 4 and is further illustrated in Figure 5. The stage of pre-processing consists of preliminary normalization and filtering. The purpose of the normalization process is to amplify the input signal to bring it to a certain level on the basis of which the setting of the parameters of the core units are adjusted. The filtering is intended to attenuate all components except the DPOAE signal as much as possible to enhance the quality of the input signal. This can be achieved by means of a simple second order band pass filter the center frequency of which is set to be that of the DPOAE signal.

The intermediate signal out of which the two artifacts are removed may be input to a third core unit for the extraction of the DPOAE signal as suggested in Figure 4. Since elimination of the two artifacts needs certain convergence time, at the very early initial moments a large portion of the two artifacts exist which will set the initial operational point of the third unit too far away from the level of the DPOAE signal. To overcome this, a time-gating process may be employed to delay the transfer of the intermediate

signal to the third unit. This is accommodated in the mid-processing unit of Figure 5. The output of this unit is zero and remains zero for a short period of time until a more or less steady state condition for the two core units is achieved. The mid-processing may also include some normalization and band pass filtering just like the pre-processing unit.

The post-processing unit consists of denormalization of the DPOAE signal and its level to restore the original values as well as some (low pass) filtering to further smooth out the estimation of the DPOAE signal and its level.

Apart from the band pass filters employed in the pre-processing and mid-processing units and low pass filters employed in the post-processing unit, low pass filters may be employed within the core units to enhance the performance of each of the three core units.

Review of the Employed Core Units

This section reviews the structure of the core units which are the building blocks of the present DPOAE measurement method. Let $u(t)$ denote a signal comprising a number of individual sinusoidal components and noise, expressed by

$$u(t) = \sum_{k=1}^{N} A_k \sin \phi_k + n(t) \tag{1}$$

where $\phi_k = \omega_k t + \theta_k$ is the total phase, and $n(t)$ denotes the total noise imposed on the signal. The objective is to find a scheme for estimating a certain component of such input signal as fast and accurate as possible; a scheme which should not be sensitive to the noise and potential time variations of the input signal. Simplicity of the structure, for the sake of practical feasibility, is desirable.

Let $\mathcal{M}$ be a manifold containing all pure sinusoidal signals defined as

$$\mathcal{M} = \{y(t,\theta) = \theta_1 \sin(\theta_2 t + \theta_3) |\ \theta_i \in [\theta_{i,\min}, \theta_{i,\max}]\}$$

where $\theta = [\theta_1, \theta_2, \theta_3]^T$ is the matrix of parameters which belongs to the parameter space

$$\Theta = \{[\theta_1, \theta_2, \theta_3]^T \mid \theta_i \in [\theta_{i,\min}, \theta_{i,\max}]\}$$

and T denotes matrix transposition. To extract a certain sinusoidal component of $u(t)$, the solution has to be an orthogonal projection of $u(t)$ onto the manifold $\mathcal{M}$, or equivalently has to be an optimum θ which minimizes a distance function d between $y(t,\theta)$ and $u(t)$, i.e.,

$$\theta_{\text{opt}} = \arg\min_{\theta \in \Theta} d[y(t,\theta), u(t)].$$

In the least squares method d is the instantaneous distance function given by

$$d^2(t,\theta) = [u(t) - y(t,\theta)]^2 \triangleq e(t)^2.$$

The error function $e(t)$ is the totality of the components present in input signal $u(t)$ other than the component of interest, plus the error incurred in the estimation process.

The parameter matrix θ is estimated by using the gradient descent method as follows:

$$\frac{d}{dt}\theta(t) = -\mu \frac{\partial}{\partial \theta}[d^2(t,\theta)]$$

where the positive diagonal matrix μ is the algorithm regulating constant. It controls the convergence rate as well as the stability of the algorithm.

Following the steps outlined above, a set of differential equations is obtained. The governing set of equations of this algorithm can be written as

$$\dot{A} = \mu_1 e \sin\phi, \tag{2}$$

$$\dot{\omega} = \mu_2 e A \cos\phi, \tag{3}$$

$$\dot{\phi} = \mu_3 e A \cos\phi + \omega, \quad (4)$$

$$y(t) = A \sin\phi, \quad (5)$$

$$e(t) = u(t) - y(t), \quad (6)$$

in which $u(t)$ and $y(t)$ are the input and output signals to the core algorithm, respectively. State variables A, ϕ and ω directly provide estimates of amplitude, phase and frequency of $u(t)$. Parameters μ_1, μ_2 and μ_3 are positive numbers which determine the behavior of the algorithm in terms of convergence speed and accuracy.

It has been shown that the dynamical system represented by the above set of differential equations possesses a unique asymptotically stable periodic orbit which lies in a neighborhood of the orbit associated with the desired component of the function $u(t)$.

In terms of the engineering performance of the system, this indicates that the output of the system, $y(t) = A \sin\phi$, will approach a sinusoidal component of the input signal $u(t)$. Moreover, time variations of parameters in $u(t)$ are tolerated by the system.

If the component of interest is specified in terms of its frequency implying a priori that the frequency is more or less known and almost fixed (such as the case of DPOAE signal and artifacts), the procedure of can be a bit further simplified. For this matter, supposing that the frequency is more or less fixed around $\omega_o = 2\pi f_o$, one can rewrite the equations as

$$\begin{aligned} \dot{A} &= \mu_1 e \sin\phi, \\ \dot{\phi} &= \omega_o + \mu_2 e A \cos\phi, \\ y(t) &= A \sin\phi, \\ e(t) &= u(t) - y(t). \end{aligned}$$

Figure 6 shows the implementation of the algorithm in the form of composition of simple blocks suitable for schematic software development tools.

Performance

Figure 7 shows the performance of the present technique in estimating the level of the DPOAE signals. The frequency of the DPOAE signal is 800 Hz. The two artifacts have a level of 1 V each. Three different measurements are made for three levels of background noise. For each case, the level of the DPOAE signal as well as the signal to noise ratio (signal meaning the DPOAE signal and the noise meaning the background noise) is specified. A high degree of noise immunity is observed. This high noise immunity not only obviates the need for sound-proof examination rooms, but also provides the way to reduce the level of the artifacts for more patient-friendly tests.

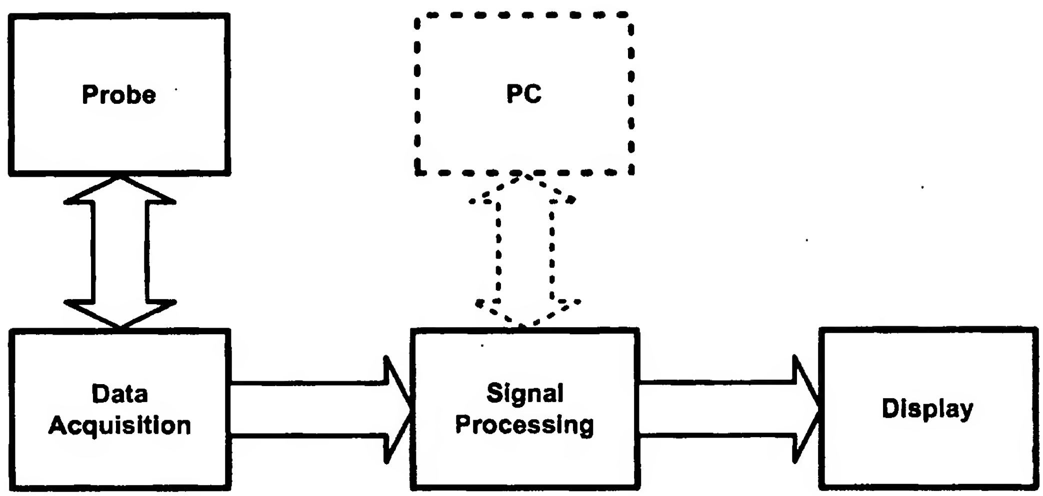

FIG. 1

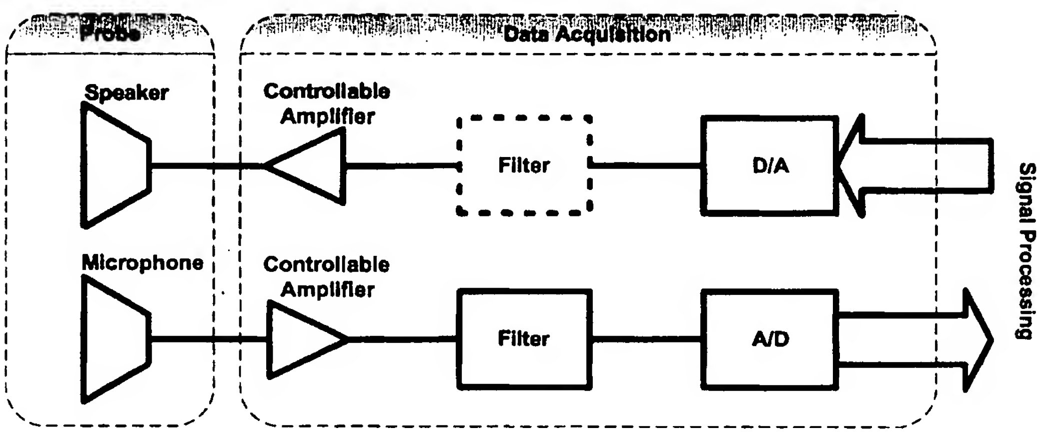
Probe
Data Acquisition
Speaker
Controllable Amplifier
Filter
D/A
Microphone
Controllable Amplifier
Filter
A/D
Signal Processing

FIG. 2

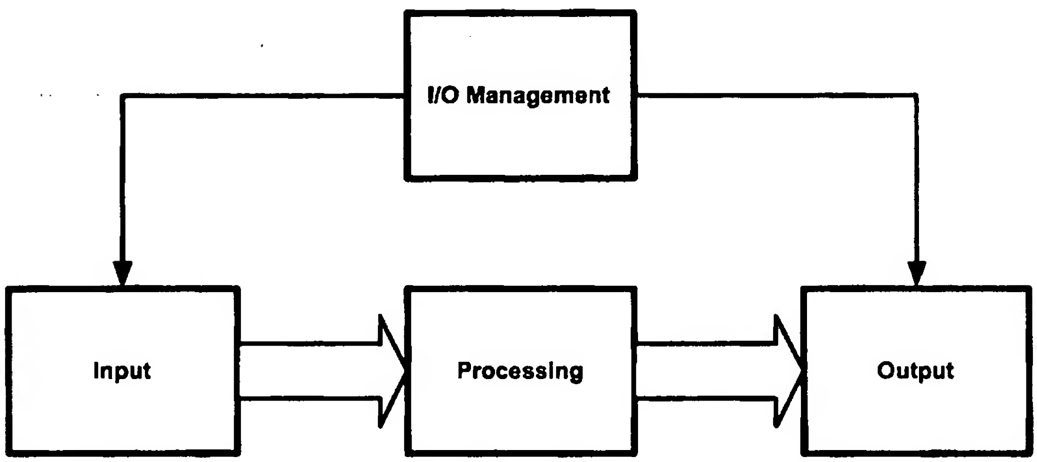

FIG. 3

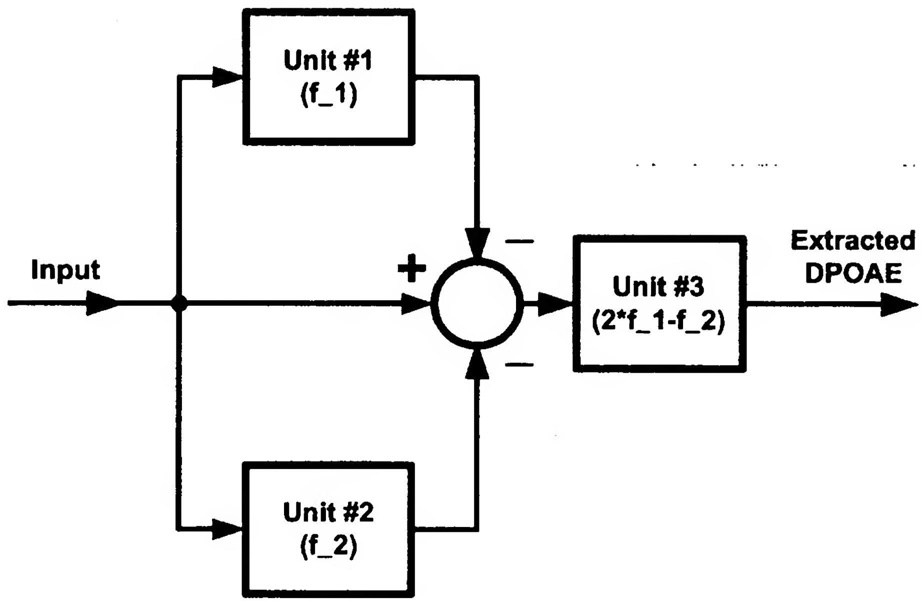

FIG. 4

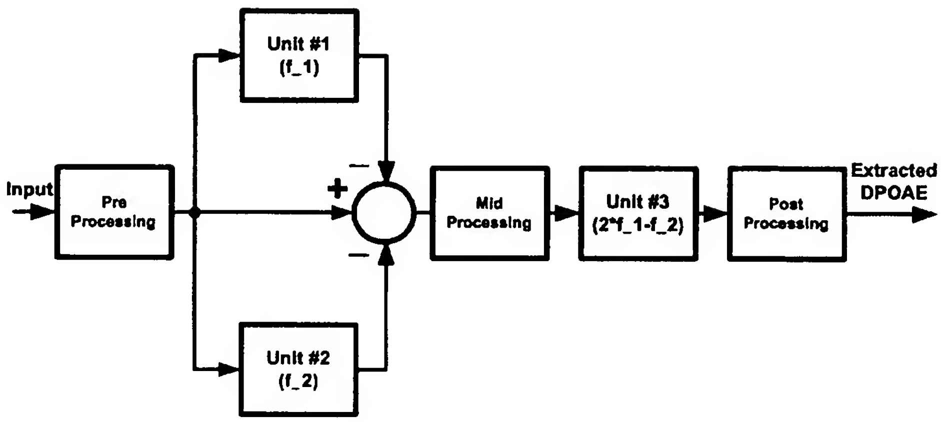
Input
Pre Processing
Unit #1 (f_1)
Unit #2 (f_2)
+
−
−
Mid Processing
Unit #3 (2*f_1-f_2)
Post Processing
Extracted DPOAE

FIG. 5

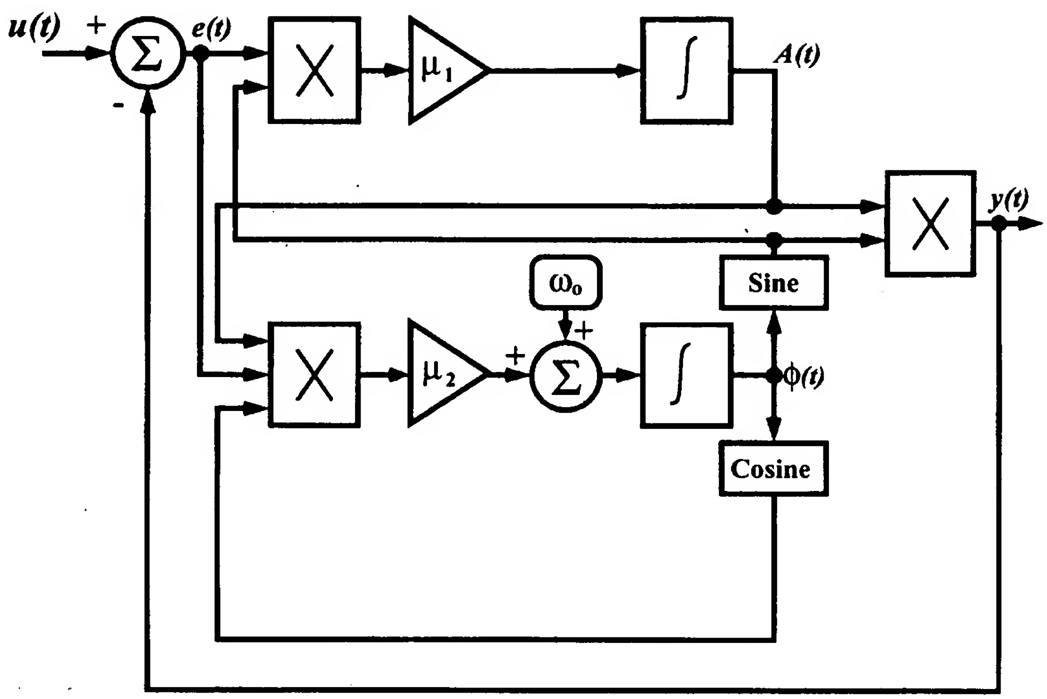

FIG. 6

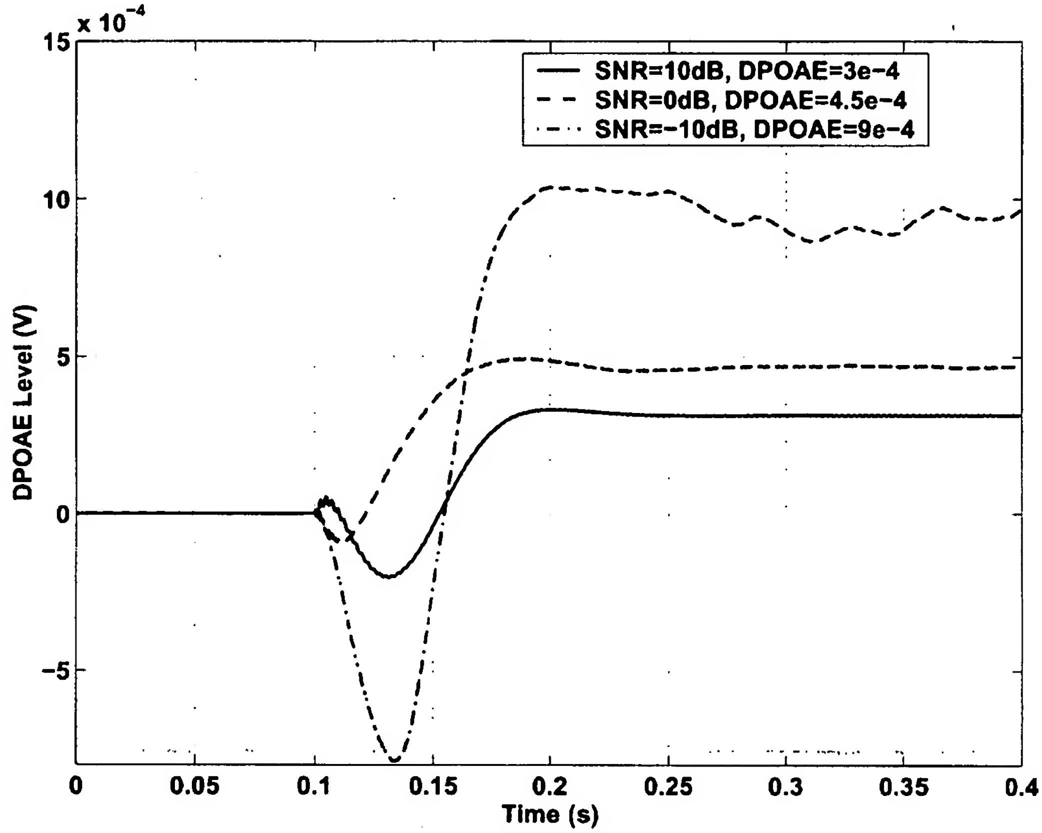
x 10^-4
15
10
5
0
-5
DPOAE Level (V)
0
0.05
0.1
0.15
0.2
0.25
0.3
0.35
0.4
Time (s)
SNR=10dB, DPOAE=3e-4
SNR=0dB, DPOAE=4.5e-4
SNR=-10dB, DPOAE=9e-4

FIG. 7